

## CLAIMS

What is claimed is:

- 5           1.       A packet switched communications system having a dynamic voice jitter buffer  
for use with voice over Internet protocol (VoIP) packets comprising:  
          a source transmitting at least one VoIP packet;  
          at least one router for routing the VoIP packet to a specified destination;  
          a destination for receiving the at least one VoIP packet; and  
10           wherein the VoIP packet operates to convey congestion information regarding the packet  
switched communications system to at least one buffer located at the destination.
2.       A packet switched communications system as in claim 1, wherein the VoIP  
packet conveys congestion information comprising the steps of:  
15           setting the time-to-live (TTL) field in the VoIP packet to a predetermined value;  
          decrementing the TTL value by one count as it traverses each respective router in the  
packet switched communications system;  
          calculating the number of routers the VoIP packet has passed through based on a final  
TTL value determined at the destination; and  
20           adjusting the capacity of the at least one buffer at the destination based on the final TTL  
value in order to mitigate non-periodic receipt of incoming VoIP packets at the destination.
3.       A packet switched communications system as in claim 2, further including the  
step of:  
25           selecting a first, second or third capacity of the at least one buffer based upon the final  
TTL value.
4.       A packet switched communications system as in claim 1, wherein the VoIP  
packet conveys congestion information comprising the steps of:

determining the speed upon which the VoIP packet has been received at the at least one router;

setting at least one field in the VoIP packet to indicate if the packet has traversed at least one previous router below a predetermined speed; and

5        adjusting the capacity of at least one buffer at the destination based upon recognition of the at least one field in order to mitigate non-periodic receipt of incoming VoIP packets at the destination.

10        5.        A packet switched communications system as in claim 4, further including the step of:

setting the least one field within the VoIP packet with a congestion value based upon the speed of the originating link.

15        6.        A packet switch communications systems claim 4, further including the step of: setting the least one field within the VoIP packet with a congestion value based upon the speed of the destination link.

20        7.        A packet switched communications system as in claim 1, further including the step of: selecting a first, second or third capacity of the at least one congestion value.

8.        A packet switched communications system as in claim 1, wherein the VoIP packet conveys congestion information comprising the steps of:

25        determining at the at least one router if a received packet has encountered at least one congested router;

setting at least one field in the VoIP packet indicating if the communications speed of a destination link is below a predetermined threshold; and

30        adjusting the capacity of at least one buffer at the destination based upon recognition of the at least one field in order to mitigate non-periodic receipt of incoming VoIP packets at the destination.

9. A packet switched communications system as in claim 8, further comprising the step of:

selecting a first, second or third capacity of the at least one buffer based upon a value set  
5 within the at least one field.

10. A method for adjusting the size of a jitter buffer for use in a voice over Internet protocol (VoIP) packet switched communications system comprising the steps of:

adjusting the time-to-live (TTL) field in a VoIP packet to a predetermined value at a  
10 source;

decrementing the TTL field by at least one count each time the VoIP packet traverse a router in the VoIP packet system;

reading the TTL field at a destination; and

adjusting the size of a jitter buffer based upon the TTL value in order to mitigate the  
15 effect of receipt of non-period VoIP packets at the destination.

11. A method for adjusting the size of a jitter buffer, as in claim 10:  
wherein the jitter buffer is located at the destination.

12. A method for adjusting the size of a jitter buffer, as in claim 10 further includes the steps of:

comparing the predetermined value of the TTL field with the value read at the destination to produce a compared value; and

mapping the compared value to a predetermined jitter buffer capacity to provide a  
25 substantially continuous flow of VoIP packets from jitter buffer.

13. A method for adjusting the size of a jitter buffer as in claim 12, further comprising the step of:

setting the capacity of the jitter buffer to either a first, second or third predetermined  
30 capacity based upon the compared value.

14. A method for adjusting the size of a jitter buffer for use with a packet network transmitting voice over Internet protocol (VoIP) packets based upon transmission path delay comprising the steps of:

5 determining the amount of transmission delay through a transmission path that a VoIP packet has encountered upon receipt by at least one router in the packet network;  
setting a field within the VoIP packet when the transmission rate for a link used for the VoIP is below a predetermined threshold;  
recognizing the field at a destination of the VoIP packet; and  
adjusting the size of a jitter buffer based upon recognition of the field in order to mitigate  
10 the effect of receipt of non-periodic VoIP packets at the destination.

15 15. A method for adjusting the size of a jitter buffer as in claim 14, wherein the jitter buffer is located at the destination.

16. A method for adjusting the size of a jitter buffer as in claim 14, further including  
the steps of:  
setting the field using a numeric value based upon the amount of transmission path delay;  
and  
mapping the numeric value into a minimal jitter buffer size required for that amount of  
20 delay.

17. A method for adjusting the size of a jitter buffer as in claim 14, further comprising  
the step of:  
adjusting the size of the jitter buffer to either a first, second or third capacity based upon  
25 the numeric value set within the field.

18. A method for adjusting the size of a jitter buffer for use with a packet network transmitting voice over Internet protocol (VoIP) packets based upon transmission path delay comprising the steps of:

5 determining the amount of transmission delay through a transmission path that a VoIP packet has encountered upon receipt by at least one router in the packet network;  
setting a field within the VoIP packet when the congestion of the link exceeds a predetermined threshold;  
recognizing the field at a destination of the VoIP packet; and  
adjusting the size of a jitter buffer based upon recognition of the field in order to mitigate  
10 the effect of receipt of non-periodic VoIP packets at the destination.

19. A method for adjusting the size of a jitter buffer as in claim 18, wherein the jitter buffer is located at the destination.

15 20. A method for adjusting the size of a jitter buffer as in claim 18, further including the steps of:  
setting the field using a numeric value based upon the link congestion; and  
mapping the numeric value into a minimal jitter buffer size required for that amount of delay.

20 21. A method for adjusting the size of a jitter buffer as in claim 18, further comprising the step of:  
adjusting the size of the jitter buffer to either a first, second or third capacity based upon the numeric value set within the field.

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